

Avaya Solution & Interoperability Test Lab

### Application Notes for the Intervoice IVR MediaServer Gateway Configuration with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

#### Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services (SES), and Intervoice MediaServer. The Intervoice MediaServer (MS) is SIP based VoIP software which provides an IVR driven-menu for executing Voice Extensible Markup Language (VXML) based applications. For the purpose of compliance testing, several demo VXML IVR applications were provided by Intervoice to exercise SIP call flows with SIP and non-SIP telephones. The Intervoice MS is configured as a trusted host in Avaya SES and a SIP trunk is established between Avaya SES and Intervoice MS.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

### 1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 4.0, Avaya SIP Enablement Services (SES) 3.1.2, and Intervoice MediaServer (MS) 3.5. The Intervoice MS is a SIP based VoIP software which provides an IVR driven-menu for executing Voice Extensible Markup Language (VXML) based applications. For the purpose of compliance test, several demo VXML IVR applications were provided by Intervoice to exercise SIP call flows with SIP and non-SIP telephones. Intervoice MS is configured as a trusted host in Avaya SES and a SIP trunk is established between Avaya SES and Intervoice MS.

**Figure 1** illustrates a sample configuration consisting of Avaya S8710 Servers, an Avaya G650 Media Gateway, an Avaya SES server, and the Intervoice MS's. Avaya Communication Manager was installed on S8710 Servers. The solution described herein is also extensible to other Avaya Servers and Media Gateways. For completeness Avaya 4600 Series SIP IP Telephones, Avaya one-X<sup>TM</sup> Desktop Edition, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones are included in **Figure 1** to demonstrate calls with the SIP-based Intervoice MS and Avaya SIP, H.323, and digital phones. The analog PSTN phone is also included to demonstrate calls routed by Avaya Communication Manager to the Intervoice MS.

The Intervoice MS is configured as a trusted host in Avaya SES. This setup does not require administration of SIP extensions in Avaya SES and Intervoice MS and avoids the overhead of periodic registrations with Avaya SES. The Intervoice MS is configured with G711 and G729 using RFC2823 for DTMF.

Typical call flows in this configuration between Avaya Communication Manager and Avaya SES, and the Intervoice MS are as follows:

- Calls originate from PSTN, H.323 or SIP trunks/endpoints to a destination number associated with the Intervoice MS.
- Avaya Communication Manager uses Automatic Alternate Routing (AAR) to route the calls to Avaya SES over SIP trunks.
- Avaya SES matches the dialed string to route the calls to the Intervoice MS.

The Intervoice MS either completes the call or transfers the call to an extension in Avaya Communication Manager by sending a REFER message to Avaya SES.



**Figure 1: Sample configuration** 

# 2. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided:

Equipment	Software/Firmware
Avaya S8710 Server	Avaya Communication Manager 4.0.0
	(R014x.00.0.730.5)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12, FW039
TN799DP C-LAN Interface	HW01, FW024
TN2302AP IP Media Processor	HW20, FW116
Avaya SIP Enablement Services Server	3.1.2 (SES-3.1.2.0-309.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323)
	2.5 (4625SW H.323)
	2.2.3 (4610SW SIP)
Avaya one-X Desktop IP Phone	R2.1 SP1
Avaya 6400 and 8400 Series Digital Telephones	-
Analog Telephone	-
Intervoice MediaServer	3.5

# 3. Configure Avaya Communication Manager

This section describes a procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES which includes steps for setting up an IP codec set, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields. These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Refer to [1] for additional details.

### 3.1. Capacity Verification

Step	Description								
1.	Enter the <b>display system-parameters customer-options</b> command. Verify that there are sufficient <b>Maximum Off-PBX Telephones</b> – <b>OPS</b> licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.								
	display system-parameters customer-options Page 1 of 10 OPTIONAL FEATURES								
	G3 Version: V13								
	Location:1RFA System ID (SID):1Platform:8RFA Module ID (MID):1								
	USED Platform Maximum Ports: 44000 1194 Maximum Stations: 36000 446								
	Maximum XMOBILE Stations: 0 0 Maximum Off-PBX Telephones - EC500: 5 0								
	<b>Maximum Off-PBX Telephones - OPS: 200 70</b> Maximum Off-PBX Telephones - PBFMC: 0 0								
	Maximum Off-PBX Telephones - PVFMC: 0 0 Maximum Off-PBX Telephones - SCCAN: 0 0								
	(NOTE: You must logoff & login to effect the permission changes.)								

Step	Description							
2.	Proceed to <b>Page 2</b> of <b>OPTIONAL FEATURES</b> form. Verify that the <b>Maximum</b> <b>Administered SIP Trunks</b> field is configured to accommodate the SIP trunk requirements. If not, contact an authorized Avaya account representative to obtain additional licenses.							
	Note: Each SIP call between two SIP endpoints (whether internal o	r external) requires						
	two SIP trunks for the duration of the call. The license file installed	on the system controls						
	the maximum permitted.	,						
	display system-parameters customer-options	Page 2 of 10						
	OPTIONAL FEATURES							
	IP PORT CAPACITIES	USED						
	Maximum Administered H.323 Trunks: 400	298						
	Maximum Concurrently Registered IP Stations: 1000	2						
	Maximum Administered Remote Office Trunks: 0 0							
	Maximum Concurrently Registered Remote Office Stations: 0 0							
	Maximum Concurrently Registered IP eCons: 0	0						
	Max Concur Registered Unauthenticated H.323 Stations: 100	0						
	Maximum Video Capable H.323 Stations: 100	12						
	Maximum Video Capable IP Softphones: 100	б						
	Maximum Administered SIP Trunks: 5000	253						
	Maximum Number of DS1 Boards with Eabo Cancellation. 0	0						
	Maximum TN2501 VAL Boards: 10	1						
	Maximum G250/G350/G700 VAL Sources: 0	0						
	Maximum TN2602 Boards with 80 VoIP Channels: 128	0						
	Maximum TN2602 Boards with 320 VoIP Channels: 128	1						
	Maximum Number of Expanded Meet-me Conference Ports: 0	0						
	(NOTE: You must logoff & login to effect the permiss	ion changes.)						

#### 3.2. IP Codec Set

This section describes the steps for administering an IP codec set in Avaya Communication Manager. This IP codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES.

Step	Description						
1.	Enter the <b>change ip-codec-set</b> < <b>c</b> > command, where <b>c</b> is a number between <b>1</b> and <b>7</b> , inclusive. IP codec sets are used in <b>Section 3.3</b> for configuring an IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, <b>G.711MU</b> and <b>G.729AB</b> were used and <b>Media Encryption</b> was set to <b>none</b> as encryption is currently not supported for SIP telephony.						
	change ip-codec	e-set 2			Page	1 of 2	
		IP	Codec Set				
	Codec Set:	2					
	Audio Codec 1: G.711MU 2: G.729AB 3: 4: 5: 6: 7:	Silence Suppression n n	Frames Per Pkt 2 2	Packet Size(ms) 20 20			
	Media Encr 1: <b>none</b> 2: 3:	ryption					

#### 3.3. IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description					
1.	Enter the <b>change ip-network-region</b> < <b>n</b> > command, where <b>n</b> is a number between <b>1</b> and					
	<b>250</b> inclusive and configure as follows:					
	• Authoritative Domain – This should match the SIP Domain value in Section 4,					
	Step 2.					
	• Intra-region IP-IP Direct Audio – Set to ves to allow direct IP-to-IP audio					
	connectivity between endpoints configured in Avava Communication Manager or					
	A vava SES in the same IP network region					
	Avaya SES in the same in network region.					
	• Codec Set – Set the codec set number as provisioned in Section 3.2.					
	• Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio					
	connectivity between endpoints configured in Avaya Communication Manager or					
	Avaya SES in different IP network regions.					
	change ip-network-region 2 Page 1 of 19					
	IP NETWORK REGION					
	Region: 2					
	Name:					
	MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes					
	Codec Set: 2 Inter-region IP-IP Direct Audio: yes					
	UDP Port Min: 2048 IP Audio Hairpinning? y					
	UDP Port Max: 65535					
	DIFFSERV/TOS PARAMETERS RTCP Reporting Enabled? y					
	Call Control PHB Value: 46 RTCP MONITOR SERVER PARAMETERS					
	Audio PHB Value: 46 Use Default Server Parameters? Y					
	802.1P/O PARAMETERS					
	Call Control 802.1p Priority: 6					
	Audio 802.1p Priority: 6					
	Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS					
	H.323 IP ENDPOINTS RSVP Enabled? n					
	H.323 Link Bounce Recovery? y RSVP Refresh Rate(secs): 15					
	Idle Traffic Interval (sec): 20 Retry upon RSVP Failure Enabled: y					
	Keep-Alive Interval (sec): 5 RSVP Profile: guaranteed-service					
	Acceptative counters a novi unescived (bbe) filb value. 10					

Step				D	escription				
2.	Proceed	Proceed to Page 3 of IP network region configuration and enable inter-region connectivity							
	between	n regions a	s per belo	w. For this co	mpliance testing	g, codec set was set to	the IP		
	codec s	et configu	red in Sec	tion <b>3.2</b> .					
	change	ip-networ	k-region	2		Page	3 of 19		
			Inte	er Network Re	gion Connection	n Management			
	src ds	t codec	direct	Total	Video		Dyn		
	rgn rg	n set	WAN WA	AN-BW-limits	WAN-BW-limits	Intervening-regions	CAC IGAR		
	2 1	2	У	NoLimit			n		
	2 2	2							
	2 3 2 4								
	2 5								
	2 6								
	2 7								
	2 8								
	29								
	2 10								
	2 13								
	2 14								
	2 15								

#### 3.4. IP Node Names

This section describes the steps for setting an IP node name for Avaya SES in Avaya Communication Manager.

Step	Description						
1.	Enter the <b>change node-names ip</b> command and add a node name for Avaya SES along with its IP address.						
	change node-name	es ip	Page 1 of 1				
	IP NODE NAMES						
	Name	IP Address					
	CLAN-1A06	192.45 .100.147					
	MEDPRO-1A13	192.45 .103.148					
	SES	192.45 .52 .160					

### 3.5. SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description					
1.	Issue the command <b>add signaling-group</b> < <b>s</b> >, where <b>s</b> is an available signaling group and configure the following:					
	• <b>Group Type</b> – Set to <b>sip.</b>					
	• <b>Transport Method</b> – Set to <b>tls</b> .					
	• Near-end Node Name - Set to CLA	N name as displayed in Section 3.4.				
	• Far-end Node Name - Set to Avaya	a SES name configured in Section 3.4.				
	• Far-end Network Region - Set to t	he region configured in Section 3.3.				
	• Far-end Domain - This should mat	ch the SIP Domain value in Section 4, Step 2.				
	• <b>DTMF over IP</b> - Set to <b>rtp-payloa</b>	<b>d</b> (RFC2833).				
	add signaling group 10 SIGNALI	NG GROUP				
	Group Number: 10 Group Type: sip					
	Near-end Node Name: CLAN-1A06	Far-end Node Name: SES				
	Near-end Listen Port: 5061 Far-end Listen Port: 5061					
	Far-end Domain:devconnect.com	Fai-end Network Region. 2				
		Demogra If ID Thursdold Tursdodd a				
		Bypass II IP Inresnoid Exceeded? n				
	DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y				
	Enable Laver 3 Test? n	IP Audio Hairpinning? N				
	Session Establishment Timer(min): 120					

### 3.6. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step		Desc	ription					
1.	Issue the command <b>add trunk-group</b> < <b>t</b> >, where <b>t</b> is an unallocated trunk group and configure the following:							
	<ul> <li>Group Type – Set to the Group Name – Enter any</li> <li>TAC (Trunk Access Code</li> <li>Signaling Group – Set to 3.5.</li> <li>Number of Members – A large enough to accommon</li> </ul>	Group Ty descriptiv e) – Set to a the Group Allowed va date the nu	pe field e name. any avai p Numb lues are umber of	value lable <b>er</b> fie betwo SIP t	configured in Section 3.5. trunk access code. ld value configured in Section een 0 and 255. Set to a value telephone extensions being used.			
	<b>Note:</b> Each SIP call between two two SIP trunks for the duration of the maximum permitted.	SIP endpo f the call. T	ints (wh The licen	ether se file	internal or external) requires e installed on the system controls			
	add trunk-group 10	TRUNK G	ROUP		Page 1 of 21			
	Group Number: 10 Group Name: SIP-SES-DevConl Direction: two-way C Dial Access? n Queue Length: 0 Service Type: tie	<b>Grou</b> Dutgoing D. Autl	p <b>Type:</b> COR: isplay? h Code?	<b>sip</b> 1 n	CDR Reports: y TN: 1 TAC: 110 Night Service:			
					Signaling Group: 10 Number of Members: 150			

#### 3.7. Dialplan/AAR/Route Pattern

This section describes the steps for setting the Dialplan, AAR digit analysis and Route Pattern in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES. These calls are ultimately destined for the Intervoice MS.

Step	Description						
1.	Issue the comm	Issue the command change uniform-dialplan <dialstring> where dialstring is the string</dialstring>					
	to match for the	to match for the dialed number, and configure as follows:					
	Matchir	ng Pattern	– Set it to a va	alue fo	or routi	ng calls to	Avaya SES for proper
	AAR dig	git analysis.					
	• <b>Len</b> – T	he dialed st	ring length to	be an	alyzed.		
	• <b>Del</b> – Se	et to <b>0</b> .					
	• Net $-S\epsilon$	et to <b>aar</b> .					
	change uniform	-dialplan	5 IINTEORM	DTAT.	י זאג.זס	LABI.E	Page 1 of 2
			UNIFORM	DIAL	FLIAN	TADUE	Percent Full: 02
	Matching		Inserted			Node	
	Pattern Len	Del	Digits	Net	Conv	Num	
	54000 5	0		aar	n n		
					n		
2.	Issue the comm	and change	public-unkr	own-	numbe	ering <e></e>	where <b>e</b> is extension code
	to be administer	ed.	P				,
	• Ext Len	- Set to the	e length of cal	lling p	artv nu	umber.	
	Ext Cod	le – Extensi	ion Code to be	e admi	nistere	d. Set to 5	in this example.
	Trk Gr	<b>p<s></s></b> - Trun	ik Group/s wh	ere th	e call i	s coming	on.
	Total C	PN Len – I	Length of the	dialed	numbe	er.	
			U				
	change public-	unknown-nu	mbering 5				Page 1 of 2
		NUMB	ERING - PUBL	IC/UNE Tot	NOWN E	ORMAT	
	Ext Ext	Trk	CPN	CI	PN		
	Len Code	Grp <s></s>	Prefix	Le	en		
	5 5	21		Ę	5		

Step		Description					
3.	Issue the command change	route-pattern <r>, where</r>	e <b>r</b> is the number of the route pattern				
	to be administered.						
	• <b>Grp No</b> – Set to the Trunk Group provisioned in <b>Section 3.6</b> .						
	• <b>FRL</b> – Set to <b>0</b> .						
	change route-pattern 10		Page 1 of 3				
	Patte	rn Number: 1 Pattern	Name: SES SIP				
	Gro FRI, NPA Pfx Hop T	SCCAN? n Secure	DCS/IXC				
	No Mrk Lmt I	ist Del Digits	QSIG				
	1. 10 0	Dgts	Intw				
	1: 10 0 2:		n user				
	3:		n user				
	4:		n user				
	5:		n user				
	0.		n user				
	BCC VALUE TSC CA-TS	C ITC BCIE Service/F	eature PARM No. Numbering LAR				
	0 1 2 3 4 W Reque	st	Dgts Format				
	1: vvvvvn n	rest	Subaddress				
	2: yyyyyn n	rest	none				
	3: уууууп п	rest	none				
	4: yyyyyn n 5: yyyyn n	rest	none				
	6: yyyyyn n	rest	none				
	* * * *						
4.	Issue the command <b>change</b>	aar analysis 5 and config	ure as follows:				
	Dialed String - Set	it to same value as <b>Match</b>	ing Pattern in Sten 1				
	Total Min and May	Set it to same value as	I on in Ston 1				
	Deute Dettern Sec	a value for a route patter	Left in Step 1.				
	• Koule Fallefii – Se	a value for a foule patient	n defined in Step 5.				
	• Call Type – Set to a	ar					
	• ANI Reqd – Set to I	1					
	Change aar analysis 5	AAR DIGIT ANALYSIS	Page 1 of 2				
			Percent Full: 2				
	Dialed	Total Route Ca	11 Node ANI De Num Regd				
	54000	<b>5 5 10 aa</b>	r n				
	2	5 5 15 aa	r n				
	2	7 7 999 aa	r n				
	245	5 5 33 aa	r n				

# 4. Configure Avaya SIP Enablement Services

This section describes the steps for configuring Avaya SES to communicate with Avaya Communication Manager and the Intervoice MS. The Intervoice MS will be configured as a trusted host with Avaya SES and a host map will be created in Avaya SES for all the calls destined for Intervoice MS. Refer to [3, 4] for additional details.

Step	Description							
1.	Open a web browser,	Open a web browser, enter http:// <ip address="" avaya="" of="" server="" ses="">/admin for the URL,</ip>						
	and log in with the appropriate credentials. Click on the "Launch Administration Web							
	Interface" link upon successful login							
	interface mix apon	successiul log						
2	On the <b>SIP Server M</b>	anagement n	age.					
	Click the Loion to expend the entione and the Comment Conference C							
	• Click the + sig	gii to expand u	ne options under Server Comiguration.					
	• Click System	Properties.						
	• Verify the <b>SIF</b>	<b>P Domain</b> mat	tches the <b>Far-end Domain</b> field value configured for					
	the signaling g	group on Avay	a Communication Manager in Section 3.5.					
			-					
	A\ /A\ /A		parts in a second se					
	AVAYA		alle Mar					
	Help Exit		Server: 192.45.52.160					
	Ten							
	Users	Tedit System F	Properties					
	Conferences	SEE Horriso	CEC.2 1 1 0-114 0					
	Media Server Extensions	System Configuration	simalex					
	Emergency Contacts	Host Type	home/edge					
	P Hosts							
	Media Servers	SIP Domain*	devconnect.com					
	Services	Note that the DNS doma	ain is: devconnect.com					
	Server Configuration	If you are unsure about domain should be the ro	this field, most often the SIP ot level DNS domain. For example.					
	System Properties	for a DNS domain of eas	stcoast.example.com, the SIP					
	Admin Accounts	allows SIP calls and inst-	ant messages to users with handles					
	License	of the format handle@example.com						
	IM Log Settings							
	SN//P Configuration License Host* Jocelhost							
	Certificate Management	Cartificate Managament						
	Trace Looper	Local IP	192.45.52.160					
	Excort/Import to ProVision	Local Name	SES-DevCon1.devconnect.com					
	Update	Logical IP	192.45.52.160					
		Logical Name	SES-DevCon1.devconnect.com					
		Gateway IP Address	192.45.52.1					

Step	Description			
3.	<ul> <li>To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a media server corresponding to Avaya Communication Manager from the SIP</li> <li>Server Management page: <ul> <li>Click the + sign to expand the options under Media Servers.</li> <li>Click Add.</li> </ul> </li> </ul>			
	AVAYA Help Exit			
	Top • Users Manage Media Server Interfaces			
	<ul> <li>Conferences</li> <li>Media Server Extensions Emergency Contacts</li> <li>Hosts</li> <li>Media Servers List Add</li> <li>Adjunct Systems Services</li> <li>Server Configuration</li> <li>Certificate Management IM Logs</li> <li>Trace Logger</li> <li>Export/Import to ProVision Update</li> </ul>			

Step	Description			
4.	At the Add Media Server Interfa for connectivity to Avaya Commu • SIP Trunk Link Type - S • SIP Trunk IP Address - • Click Add when finished	ace page, provision SIP Tru- unications Manager: Set to the Transport Metho Set to the CLAN IP address and then click Continue on	unk parameters as follows od field value in Section 3.5. as displayed in Section 3.4. the confirmation page (not	
	shown).			
	Top Users Conferences Media Server Extensions Emergency Contacts Hosts Media Servers List Add Adjunct Systems Services Server Configuration Certificate Management IM Logs Trace Logger Export/Import to ProVision Update	Add Media Server Media Server Interface Name* Host SIP Trunk SIP Trunk Link Type SIP Trunk IP Address* Media Server Media Server Admin Address (see Help) Media Server Admin Login Media Server Admin Password Media Server Admin Password Confirm Fields marked * are required.	Interface         \$8710         192.45.52.160 ▼         ○ TCP ● TLS         192.45.100.147	

Step	Description				
5.	A Host Address Map is requ	A Host Address Map is required on Avaya SES to direct outbound calls from Avaya			
	<ul> <li>Communication Manager to the Intervoice MS. An Address Map is used to route the calls based on the contents of SIP INVITE URI. To configure Host Address Map, do the following:</li> <li>Click the + sign to expand the options under Hosts.</li> </ul>				
Click Add Another Map.					
	ΑνΑγΑ				
	Help Exit				
	Top				
	Users	📕 List Host Add	ress Map		
	Conferences				
	Media Server Extensions	Host 192.45.52.160			
	Emergency Contacts	Commands Name	Commands Con	tact	
	= Hosts	Add Another Map	Add Another Contac	t Delete Group	
	Update All				
	List	Add Map In New Group			
	Migrate Home/Edge				
	Media Servers				
	Adjunct Systems				
	Services				
	Server Configuration				
	Certificate Management				
	IM Logs				
	Trace Logger				
	Export/Import to ProVision				
	Update				

Step	Description				
6.	On the Add Host Address Map page enter the following:				
	• Name – Any descriptive name.				
• <b>Pattern</b> – Expression to match the beginning of the URI.					
	• Click <b>Add</b> and then <b>Con</b>	tinue on the next page (not shown).			
	AVAYA				
	Help Exit				
Top Users Add Host Address Map					
	<ul> <li>Users</li> <li>Conferences</li> <li>Media Server Extensions Emergency Contacts</li> <li>Hosts         <ul> <li>Update All</li> <li>List</li> <li>Migrate Home/Edge</li> </ul> </li> <li>Media Servers</li> <li>Adjunct Systems Services</li> <li>Server Configuration</li> <li>Certificate Management IM Logs</li> <li>Trace Logger</li> <li>Export/Import to ProVision</li> </ul>	Host 192.45.52.160 Name* MSTRUNKING Pattern* ^sip:54000 Replace URI I Fields marked * are required.			

Step	Description				
7.	The host contact that must be en	tered for the Address Map defined in <b>Step 7</b> and is			
	configured as follows:				
	Click Add Another Contact on the List Host Address Map screen displaye     Step 6.				
	• <b>Contact</b> – Enter the dest	ination IP address ( <i>ip_addr</i> ), port number ( <i>port</i> ) and			
	transport protocol ( <i>proto</i>	col) as follows:			
	sip:\$(use	er)@1p_addr:port;transport=protocol.			
	The IP address correspon	er)@192.45.52.201:5000;transport=5000 is entered.			
	Click on Add and then cl	lick <b>Continue</b> on the next page (not shown)			
	• Check on Add and then e	tek continue on the next page (not shown).			
	AVALYA				
	Help Exit				
	Тор				
	Users	Add Host Contact			
	Conferences				
	Media Server Extensions	Host 192.45.52.160			
	Emergency Contacts	Handle MSTRONKING			
	E Hosts	Contact* [sip:\$(user)@192.45.52.201;5060;transport=			
	Update All	Fields marked * are required.			
	List				
	Migrate Home/Edge	Add			
	Madia Servers				
	Adjunct Systems				
	Convices				
	Certificate Management				
	IM Logs				
	Trace Logger				
	Export/Import to ProVision				
	Update				

Step	Description				
8.	The IP Address of the Intervoice MS must be configured as a trusted host on Avaya S As a trusted host, Avaya SES will not issue SIP authentication challenges for incomin requests from the IP address of Intervoice MS.			Avaya SES. incoming	
	<ul> <li>The following steps are red</li> <li>Connect to Avaya S</li> <li>Issue the trustedhost or trustedhost text']. In the and SES-IF</li> <li>Important Step: C Avaya SES adminition Top screen.</li> </ul>	quired to configure SES using secure sh st command at the I t -a <i>MediaServer-II</i> is example <b>MediaS</b> P-address is set to 1 Complete the trusted stration web page a	a trusted host in Avaya SES: nell and log in using proper cred Linux shell prompt. <i>P-address</i> -n <i>SES-IP-address</i> [ - erver-IP-address is set to 192. 92.45.52.160. I host configuration by returning nd click on Update in the left p	entials. c 'comment 45.52.201 g to the ane on the	
	AVAYA Help Exit				
	Top Users	🛃 Тор			
	<ul> <li>Conferences</li> <li>Media Server Extensions</li> </ul>	Manage Users	Add and delete Users.		
	Emergency Contacts	Manage Conferencing	Add and delete Conference Extensions.		
	Media Servers     Adjunct Systems	Manage Media Server Extensions	Add and delete Media Server Extensions.		
	Services	Manage Emergency Contacts	Add and delete Emergency Contacts.		
	<ul> <li>Server Configuration</li> <li>Certificate Management</li> </ul>	Manage Hosts	Add and delete Hosts.		
	IM Logs	Manage Media Servers	Add and delete Media Servers.		
	<ul> <li>Trace Logger</li> <li>Export/Import to ProVision</li> </ul>	Manage Adjunct Systems	Add and delete Adjunct Systems.		
	Update Manage Services Start and stop server processes on this host.				
9.	Repeat <b>Step 7</b> and <b>Step 8</b> example configuration, <b>19</b> Intervoice MS.	using the IP address 2.45.52.101 was use	s of the second Intervoice MS. The address of the redur	In this idant	

### 5. Configure the Intervoice MediaServer

This section describes the steps for configuring the Intervoice MS to communicate as a trusted host with Avaya SES. This section assumes that the Intervoice MS software and Media Control Center configuration utility are already installed and IP addresses are set. Configuration steps described in this section apply only to the fields where a value needs to be modified or entered. Default values are used for all other fields. Only relevant screens and configuration steps are shown here. For further details refer to Intervoice MS documentation.

Note: Due to the page size, only the most relevant fields have been included in the screen shots.

Step	Description			
1.	Open a web brow	Open a web browser, enter <u>http://<ip address="" mediaserver="" of="">:8074/ccportal/</ip></u> for the		
	URL, and log in v	URL, and log in with the appropriate credentials.		
2	Select the System	View button on the top, click <b>Preferences</b> and click <b>Add Node</b>		
4.	Sciect the System	<b>The</b> would be the top, ener <b>Treferences</b> and ener <b>Aud Toue</b> .		
	File Edit View Eavorites To	portal/portal/media-type/ntml/role/admin/page/default.psml/js_pane/ - Microsoft Internet Explorer		
	A Back + → + (A) (A) (A)	no rop @Search © Favorites @Media Ø ■ ■		
	Address Addres	godadu i miratokos gonala go biji goda i go biji goda i goda		
	Hudi C33 C 11(tp.)//192.43.32.201.0			
	intervoice	iviadmin Last Login Time: 6/19/07 91:28:25 AM		
		About   Node View   System View   Logout   Edit Account		
	System View Preferen	nces		
	Configure Systems	View/Edit Configuration ?		
	Configure Node Types	Name Primary Type Actions		
	View/Edit Node Types	Unassigned Partitions		
	Configure Services	Unassigned Nodes		
	View/Edit Services	Add System Add Partition Add Node		
	Configure Portal			
	Configure Time Restriction			
	View Portlets			
	Properties	R. C.		
	Configure Thresholds			
	Purge/Backup Database			
	Control Database			
	Configure Database			
	Configure Control Center			
	Configure Bean Server			
	View License			
	View Audit Log			
	Ouery Audit Log History			
	Service Providers			
	View/Edit Service			
	Providers			
	View/Edit PALs			
	E Done	👘 Internet 🥼		

Step	Description				
3.	On the Add Node scre	en, configure as follows:			
	Node Name –	• Node Name – Enter any descriptive name			
	Node ID Set	to the ID address configured for Intervoice MS			
	• Node IP – Set	to the IP address configured for intervoice wis.			
	• Click <b>Submit</b> .				
	Add Nadas - Missasaß Tataunat Eurlavau				
	Add Nodes - Microsoft Internet Explore				
	The Edit view Pavorites Tools Help				
	Address en http://192.45.52.201:8070/ccpor	tayportaymedia-type/ntmiyroie/adminypage/default.psmi/eventSubmit_doShownodeadd=add&ys_T (2 40 unks * 😏 Shagit 🛐 🦷 🖓 •			
	intervoice	iviadmin Last Login Time: 6/19/07 9:28:25 AM			
		About   Node View   System View   Logout   Edit Account			
	System View Preferences				
	Configure Systems	Add Node ?			
	View/Edit Configuration				
	View/Edit Node Types	Node Name: MS1			
	Configure Services	Node IP: 192.45.52.201			
	View/Edit Services	Partition: Unassigned			
	Configure Portal				
	Configure Time Restriction	Available Assigned			
	Configure Portal				
	Properties	Add			
	Configure Thresholds	Remove			
	Control Database				
	Configure Database				
	Configure Control Center	Advanced			
	Configure Bean Server	Submit Cancel			
	View License				
	Activity Logging				
	Ouery Audit Log				
	Service Providers				
	View/Edit Service				
	Providers View/Edit BALs				
	VIEW/EUIC PALS	×			
1					

Step	Description			
4.	<ul> <li>At the System: All screen, configure as follows:</li> <li>Select the Node View on the top and then select Configure.</li> </ul>			
	• Click Add in the Configurations line to display the node configured in Step 3.			
	<ul> <li>Click Add Product and select the Media Server 3.5 from the list of choices.</li> <li>Click Submit.</li> </ul>			
	Configuration Overview - Microsoft Internet Explorer  Elle Edit View Exweiter Tools Help			
	↓ Back • → • ③ ③ ☆ ② Search ⓐ Favorites ④ Media ③ ▷ • ④ ⊠ 目			
	Address 🕘 http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/nodeview.psml	💌 🔗 Go 🛛 Links 🎽 🈏 SnagIt 🔠 👘 🔹		
	intervoice System: All	iviadmin Last Login Time: 6/28/07 11:27:26 AM Invalid Logon Attempts: 0 About Node View View Logout   Edit Account		
	All Servers Configure Manage			
	Configuration Overview	· ?		
	Nama	Actions		
	Global Configurations	Actuuis		
	CCXML-AppServer Mapping	Edit		
	Active	Actions		
	Configurations	Add		
	Ms1 (192.45.52.201)	Add Product Delete		
	Media Gateway Media Server 35 Media Server			
	Submit	_		
		-		
	ê	Internet		

Step	Description		
5.	Only one screen shown for following steps:		
	• Click + in the MS1 line to expose the Media Server 3.5 line.		
	<ul> <li>Click + in the Modio Server 3.5 line to evenese Telephony Care line</li> </ul>		
	• Check + in the Wieula Sel ver 5.5 line to expose Teleph		
	• Click Add Type and select Telephony Core from the	list of choices.	
	• Click Add and the next line with New Config Description appears.		
	<b>Note</b> : This step is repeated for all the features being configured like Telephony Core, HAL HMP Configuration, etc.		
	🚰 Configuration Overview - Microsoft Internet Explorer		
	File Edit View Favorites Tools Help	4 <u>B</u>	
	↔ Back + → - ③ ② 집   ③ Search ⓐ Favorites ④ Media ③ [ ⓑ + ④ ☑ ]		
	Address Childuration Uverview	C Go Links " Snaglt E' 2	
	Global Configurations		
	CCXML-AppServer Mapping	Eun	
	Active	Actions	
	Configurations	Add	
	⊡ 🖳 MS1 (192.45.52.201)	Add Product Delete	
	🗎 🛄 Media Server 3.5	Add Type Delete	
	🖓 🛄   Telephony Core 🖉	Add Delete	
		Add Delete	
	HAL HMP Contiguration	Add Delete	
	En la voir right for the second secon	Add Delete	
		Delete	
	E Contig Description	Add Delete	
	Application Routing     Interphony Core		
	Supplementary Services	<u>→ →</u>	
	HAL I DM Configuration		
	Call Control Configuration Submit Revert		
	VXML Browser Configuration		
	Application Routing		
	PNET Configuration		
	ê	💕 Internet	

Step		Description		
6.	Select <b>Port to Lin</b> follows: • At the <b>Boa</b> • At the <b>Boa</b> • <b>Total phys</b>	e Mapping in the left pane rds line, click Add Board rdID line, click Add Spar sical Ports – Set to a value	and <b>BoardID</b> line appendent and the next three field depending upon available	en and configure as ears. ls appear. ble licenses.
	CIICK SUDI     MICS Server Configuration - Micro     File Edit View Favorites Tools     → Back      →      →      ☆ Back     ↔      ☆ Back     ☆      ☆	MIIT. osoft Internet Explorer Help earch 🝙 Favorites ③Media 🔇 🖓 🎝 🗐 🔄 ccportal/portal/media-type/html/role/admin/page/fmtcs.psml/js;	pane/P-10930f919a3-100e7	_ B X ∰ ∂Go Links <sup>20</sup> Snagit 🛃 •
	intervoice Configure	System: All	iviadmin Las <b>About</b>   <u>Node View</u>   <u>Syste</u>	t Login Time: 6/19/07 9:28:25 AM A Invalid Logon Attempts: 0 m View   Logout   Edit Account
	Port To Line Mapping Sinared Directory	Port To Line Ma	Submit       Revert	Back ? Actions Add Board Add Span Delete Delete
				, , , , , , , , , , , , , , , , , , ,

Step	Description		
7.	Select the Active radio button associated with the Telephony Core feature and click		
	Submit to activate.		
	Note: This step is repeated for all the configured features like	Telephony Core, HAL HMP	
	configuration etc		
	configuration, etc.		
	🗿 Configuration Overview - Microsoft Internet Explorer	_ 8 ×	
		(B)	
	⇔Back • ⇒ • 🕲 😰 🚮 🔞 Search 🚵 Favorites 🛞 Media 🕉 🔄 • 🎒 🖾 🖹		
	Address 🗿 http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/nodeview.psml/js_pane/P-1134460e206-10000	🗾 🧬 Go 🛛 Links 🎽 🌀 SnagIt 🖆 👘 🔹	
	intervoice System: All	iviadmin Last Login Time: 6/27/07 4:13:51 PM Invalid Logon Attempts: 0	
	All Corriers Configure Manage	le View   <u>System View</u>   <u>Logout</u>   <u>Edit Account</u>	
	Air servers Coningure Manage		
	Configuration Overview	?	
		·····	
	Global Configurations	F-414	
	CUXML-AppServer Mapping	Eur	
	Active	Actions	
	Configurations	Add	
	Ē · 💭 MS1 (192.45.52.201)	Add Product Delete	
	🖻 🦲 Media Server 3.5	Add Type Delete	
	E Contraction Core	Add Delete	
	C E New Config Description	Edit Copy Delete	
	HAL HMP Configuration	Add <u>Delete</u>	
	C 🗄 🗐 New Config Description	Edit Copy Delete	
	Call Control Configuration	Add <u>Delete</u>	
	C 🗄 🔄 New Config Description	Edit Copy Delete	
	Application Routing	Add Delete	
	C 🗈 🗐 New Config Description	Edit Copy Delete	
		_	
	Submit Revert		
	Configured node may not operate properly because not all configurations have been applied. Select t apply your configurations.	he Active radio button then click Submit to	
	8	💌 👸 Internet	
8.	Repeat Step 5 but select HAL HMP Configuration.		
0.			



Step		Descr	ription	
10.	Select <b>Board</b> in the left pane of the <b>HMP Board</b> screen and configure as follows:			
	• IP Address – Set to the IP Address of the Intervoice MS			
	Protocol Name – S	Set to <b>SIP</b> .		
	• Click <b>Submit</b> .			
	HAL HMP EDIT BOARD - Microsoft Internet Explore File Edit View Favorites Tools Help	r		
	부 Back • => · ⑧ 환 삶 ⓒ Search  Favorit	ies 🛞 Media 🧭 🛃 🎒 🗾 🗐		
	Address 🙋 http://192.45.52.201:8070/ccportal/portal/med	dia-type/html/role/admin/page/childhmp?boar	rdId=0&cfgName=New%20Config%20Description&c 💌 🄗 G	o Links » 🌀 SnagIt 🖭 📆 🗸
	intervoice	System: All	iviadmin Last Login	Time: 6/19/07 9:28:25 AM
	HMP Board			
	Configurations		Edit Board - 0	Back ?
	Board VOLP Registration			
	Alarm	Board ID		
		Protocol Name	SIP	
			Submit	
				intervoice
				Intervoice
				<b>V</b>
	E Done			🔰 🔮 Internet

Step		Descr	iption	
11.	Select <b>Codec</b> in the left pane of the <b>HMP Board</b> screen and configure as follows:			
	<ul> <li>Click Add at the next screen (not shown).</li> <li>Codec Family – Set to the Audio Codecs field value configured in Section 3.2</li> </ul>			
				onfigured in <b>Section 3.2</b> .
	• <b>Type</b> – Se	t to the Audio Codecs field	value configured	in Section 3.2.
	Frame Size	ze – Set to the Packet Size	field value configu	red in Section 3.2
	Repeat abo	ove steps to add additional	rodecs	
	Click Sub	mit		
	• CIICK SUD	mit.		
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	File Edit View Favorites Tools	Help		
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	Address     http://192.45.52.201:8070	//ccportal/portal/media-type/html/role/admin/page/childhmp.psml	ljs_pane/P-1070b33dE1re1e7a4-3301	
	intervoice	System: All	About Node Vi	Invalid Logon Attempts: 0 ew System View Logout Edit Account
	HMP Board			
	Configurations		Codec - Board 0	Back?
	Board Board VOIP Registration		Codec Information	
	Codec	Codec Family Type	rame Size	Frames per Packet Actions
		G711Codecs G711M	20 🗸	1 Delete
		G711Codecs		Add
		G723Codecs G729Codecs	Submit	
				intervoice
				intervoice
	8			Internet

Step		Descripti	ion			
12.	Select DTMF Paylo	ad and Fax from the left pa	ne of the HALHM	<b>P</b> screen and		
	configure as follows:					
	• DTMF Dete	• <b>DTMF Detect Scheme</b> – Set to match the DTMF scheme used in Section 3.5				
	DTMF Pavl	<ul> <li>DTME Detect benefice - Set to match the DTME benefice used in Section 3.5.</li> <li>DTME Dayload - Set to 127</li> </ul>				
	• DINIT Tayl	Schome Set to Deinwite T	10			
	• Fax Delect S	scheme – set to kenivite 13	<b>90</b> .			
	🗿 VXML Properties - Microsoft Internet E	xplorer				
	File Edit View Favorites Tools Help	, ,				
	🗢 Back 🔹 🤿 🖉 🙆 🖓 🥘 Search	🛚 🔝 Favorites 🛞 Media 🛛 🛃 🚽 🎒 🗾 📃				
	Address 🙆 http://192.45.52.201:8070/ccport	tal/portal/media-type/html/role/admin/page/halhmp.psml/js_pane/l	0-1070b331e17a4-3301	🖌 🔗 Go 🛛 Links 🎽 🌀 SnagIt 😁 🧉	<u>₽</u> -	
	intervoice	System: All	iviadmin L. <u>About</u>   <u>Node View</u>   <u>Sys</u> t	ast Login Time: 6/19/07 9:28:25 AM Invalid Logon Attempts: 0 <b>tem View</b>   <mark>Logout</mark>   <mark>Edit Account</mark>	~	
	HALHMP					
	Configurations	DTMF Payload & Fax	(New Config Description	n) <u>Back</u>	?	
	DTMF Payload & Fax			_		
	System VOIP Parameters	DTMF Detect Scheme	RFC2833			
	System VOIP Configuration	DTMF Payload	127			
		Fax Detect Scheme	Reinvite I 38			
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Step		Des	scription	
15.	Select Telephony	from left pane of the Ca	Il Control Configuration scree	en and configure
	as follows:			
	Click Dele	te at the next screen. [No	ot Shown]	
	Select Vol	<b>P</b> for the Sorver Type f	and click Submit	
		I for the Server Type h	leiu and chek Sublint.	
	🚰 Telephony - Microsoft Internet Ex	plorer	Ν	
	File Edit View Favorites Tools	Help	14	
	↔ Back • → • ③ 🙆 🖓 🗔 S	iearch 🗟 Favorites 🎯 Media 🧭 🗟 🖬 🗐		
	Address 🙆 http://192.45.52.201:8070/	ccportal/portal/media-type/html/role/admin/page/fcc?cfgN	lame=New%20Config%20Description&cfgType=CCNTRL8 🗾 🔗 Go Lir	nks » 😏 SnagIt 🛃 🔹
	intervoice	System: All	iviadmin Last Login Time: Inva <u>About</u>   <u>Node View</u>   <u>System View</u>   <u>Lo</u>	6/19/07 9:28:25 AM lid Logon Attempts: 0 gout   Edit Account
	Configure			
	Call Control Configuration	Telepho	ony - New Config Description	Back ?
	Transfer Connect		Actions	
	Outbound Configuration		Actions	
	Outbound Calling	Server Type VolP	Add Line Group	
	Route Definitions	TDM		
	CTI Configuration	VolP		
	СТІ			
			Pulumit Devert	
			Submit	
				<b>•</b>
	é			Niternet
16.	Repeat Step 8 and	but select Call Control	Configuration.	
	Repeat Step 5 but	select Application Rou	ting.	

Step	Description		
17.	Select <b>Rules</b> from left pane of the <b>Route Table</b> screen and configure as follows:		
	• Click <b>Create Rule</b> on the next screen (not shown).		
	• Application Type – Set to the type of Application to be used. Set to		
	application/voicexml+xml, a VXML application in this example.		
	• Application URL – Physical location of the VXML application on the Intervoice		
	MS.		
	Click Submit		
	🚰 Route Table Rules - Microsoft Internet Explorer		
	File Edit View Favorites Tools Help		
	→ Back + → → → 😳 🙆 💁 🔞 Search 🗟 Favorites 🤯 Media 🧭 🖏 + 🎒 🖾 📃		
	iviadmin Last Login Time: 6/19/07 9:28:25 AM		
	intervoice System: All Invalid Logon Attempts: 0 About Node View System View Logout Edit Account		
	Configure		
	Route Table Create Rule - New Config Description Back?		
	Rule ID		
	Submit Cancel		
	intervoice		
18.	Repeat Step 7 but select Application Routing.		

# 6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily to exercise the Intervoice MS IVR solution using DTMF. Tests were done to verify that the Intervoice MS is able to recognize DTMF commands and take appropriate action for calls originating from SIP, H.323, digital, analog and PSTN phones using Avaya SES and Avaya Communication Manager.

### 6.1. General Test Approach

The general test approach was to place calls from any phone to establish a call into the Intervoice MS and exercise the supported features. The main objectives were to verify that:

- The Intervoice MS successfully initiates and terminates calls to IP and non-IP telephones.
- The Intervoice MS successfully executes a blind transfer.
- The Intervoice MS successfully shuffles for VoIP calls.
- The Intervoice MS successfully handles DTMF.
- The Intervoice MS successfully handles T.38 fax.
- The Intervoice MS successfully handles signaling and RTP traffic on separate Ethernet cards.
- The Intervoice MS successfully prioritizes traffic.

For serviceability testing, failures such as cable pulls and hardware resets were applied.

#### 6.2. Test Results

The test objectives of Section 6.1 were verified. For serviceability testing, the Intervoice MS operated properly after recovering from failures such as cable disconnects, and resets of the Intervoice MS, the Avaya SES server, and Avaya Communication Manager. For redundancy testing, the calls were successfully handled by the redundant Intervoice MS. Calls placed into the Intervoice MS were successfully shuffled.

The following observations were made during testing:

- The Intervoice MS operates only with UDP as the SIP transport protocol.
- The Intervoice MS supports only Layer-3 QoS parameters.
- When calls were made using H.323 IP telephones, only the following scenarios were supported:
  - In-band DTMF with any codec
  - RTP-Payload DTMF with G729

Intervoice Inc. expects to resolve the above observations in future releases.

# 7. Verification Steps

Step	Description		
1.	Verify all members for the SIP trunk group provisioned in Section 3.6 are in-service/idle.		
	From a SAT session:		
	• Issue the command "status trunk 10".		
	• Verify that all members in Trunk Group 10 are <b>in-service/idle</b> .		
2.	Verify that the Intervoice MS is properly configured as a trusted host using secure shell to		
	connect to Avaya SES server.		
	SES> trustedhost -L		
	Third party trusted hosts.		
	Trusted Host IP address   SES Host IP address   Comment		
	192.45.50.201   192.45.53.201   MS1		
3.	Place calls into the Intervoice MS and verify that it responds with an announcement and it		
	accepts the DTMF digits and takes action based upon the number entered.		

### 8. Support

For technical support on Intervoice Inc., consult the support pages at <u>http://www.intervoice.com/support</u> or contact Intervoice Inc. technical support at:

- Phone: 1-972-484-1000
- E-mail: <a href="mailto:support@intervoice.com">support@intervoice.com</a>

# 9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 4.0, Avaya SIP Enablement Services (SES) 3.1.2, and Intervoice MediaServer 3.5. The Intervoice MediaServer (MS) is SIP based VoIP software which provides an IVR driven-menu for executing Voice Extensible Markup Language (VXML) based applications. For the purpose of compliance testing, several demo VXML IVR applications were provided by Intervoice to exercise SIP call flows with SIP and non-SIP telephones. The Intervoice MS is configured as a trusted host in Avaya SES and a SIP trunk is established between Avaya SES and Intervoice MS. The compliance testing was successful with the exception of the issues noted in **Section 6.2**.

# 10. Additional References

Product documentation for Avaya products may be found at <u>http://support.avaya.com</u>.
[1] *Administrator Guide for Avaya Communication Manager*, Issue 3.1, February 2007, Document Number 03-300509
[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 12, February 2007, Document Number 555-233-504

[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 7, May 2007, Document Number 555-245-206

[4] *Installing and Administering SIP Enablement Services R3.1.2*, Issue 4, May 2007, Document Number 03-600768

Product documentation for Intervoice Inc. products may be found at http://www.intervoice.com.

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